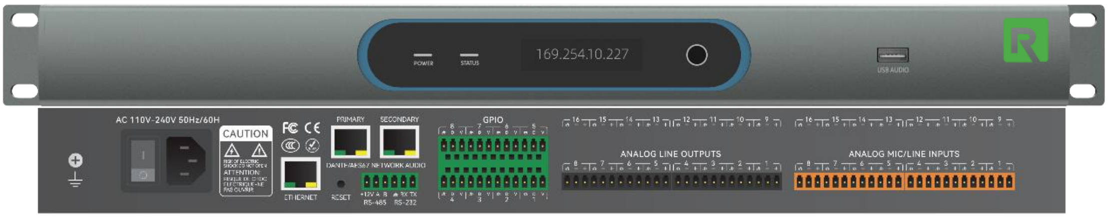
RESOUNDIFY

VoxNova88

Open Structure DSP 8X8, Dante 64X64, 8AEC



PRODUCT OVERVIEW

The **Resoundify VoxNova88** is a high-performance, open-architecture audio DSP processor tailored for professional AV and conferencing applications. It features 8x8 analog inputs and outputs, and supports Dante™ digital audio networking with 64x64 Dante I/O, enabling seamless integration across large-scale networked audio systems.

The VoxNova88 represents the next generation in flexible, scalable DSP solutions for high-end audio environments. Built with a powerful processing core and an open architecture design, it empowers AV system integrators to tailor audio processing chains precisely to the needs of any space—whether it's a compact huddle room or a large conference environment.

KEY FEATURES

- **Professional SHARC DSP Core:** Powered by the Analog Devices SHARC platform, delivering ultra-fast parallel processing in a semi-open architecture environment for custom DSP design and signal chain optimization.
- **High-Quality Audio Processing:** 24-bit/48kHz audio resolution ensures crystal-clear sound quality across all channels.
- **Intelligent Feedback Suppression:** Independent adaptive feedback suppression on each channel automatically eliminates unwanted noise.
- **Full-Duplex AEC & ANC:** Integrated Adaptive Echo Cancellation and Active Noise Cancellation for clear communication in conferencing environments.
- **Auto Mixer & Gain Control:** Built-in Gain Sharing Auto Mixer, Automatic Gain Control (AGC), and Audio Ducking (Ducker) for seamless level balancing.
- **Ambient Noise Compensation:** Real-time Ambient Noise Compensator (ANC) adjusts audio levels based on environmental sound.
- **Comprehensive Audio Matrix:** Flexible mixing matrix with input level control, channel duplication, linking, and grouping.
- **Expandable Control Options:** 8 configurable GPIOs (input/output/ADC), RS-232 & UDP support with assignable ports for central control systems.
- **Multi-Platform Compatibility:** Supports both iOS and Windows OS with dual USB audio interface for recording and conferencing.

APPLICATIONS

- Boardrooms
- Classrooms
- Auditorium

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TECHNICAL SPECIFICATIONS

System Specifications

Processor	ADI SHARC 21569@1GHz SIMD*2
Raw Processing Capacity	500 MIPS, 6 GFLOPS, 2 GMACS
Sampling Rate	48 kHz ± 100 ppm
Frequency Response (A/D/A)	20 Hz - 20 kHz ± 0.3 dB
Dynamic Range (A/D/A)	115 dB (A-weighted)
THD + Noise	<0.003%@4dBu
Channel Separation (A/D/A)	110 dB @ 1 kHz, +24 dBu
Latency (A/D/A)	<4 ms (input routed directly to output)
Delay Memory	174 mono seconds
Analog Control Inputs	0-3.3 VDC
Recommended External Control Potentiometer	10k Ohm, linear taper
Logic Outputs	Low (0 V) when active Pulled high (5 V) when inactive
Logic Output Maximum External Power Supply / Current Sinking	24 VDC / 50 mA
Logic Output Maximum Output Current	10 mA
RS-232 Accessory Serial I/O	57.6 kbps (default), 8 data bits, 1 stop bit, no parity, Straight-through wiring; pins 2, 3, 5 used
Maximum Stored Presets	1,000 storable presets

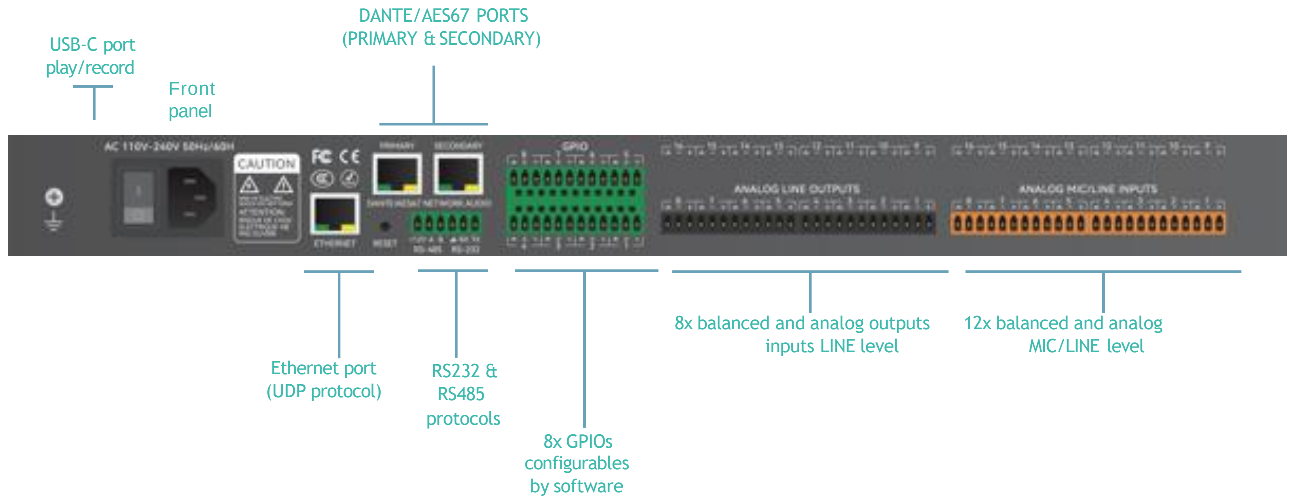
Analog Inputs and Outputs

Number of Analog Inputs	8 switchable balanced mic or line level
Analog Input and output Connectors	3.81 mm terminal blocks
Nominal Analog Input and output Level	+4 dBu with 20 dB headroom
Analog Input and output Maximum Level	+24 dBu (or +22.8 dBu into a 2k Ohm minimum load)
Analog Mic Pre-amp Gain	0 to 51 dB (in 3 dB steps) with ±24 dB digital trim
Analog Mic Pre-amp EIN	< -125 dB (with 150 Ohm source, 22.4 kHz BW)
Analog Input Impedance	5.4k Ohms balanced, 3k Ohms unbalanced
Analog Phantom Power (per input)	+48 VDC per input, max 10 mA
Analog Input Dynamic Range	>115 dB, A-weighted
Analog Input THD + Noise	<-100 dB (22.4 kHz BW, unweighted), 1 kHz @ +15 dBu, 0 dB gain
Analog Input Latency	2.5 ms
Number of Analog Outputs	8 balanced line level
Analog Output Impedance	600 Ohms balanced, 300 Ohms unbalanced
Analog Output Dynamic Range	115 dB, A-weighted
Analog Output THD + Noise	< -97 dB (22.4 kHz BW, unweighted); 1 kHz, 0 dB gain, +8 dBu output
Analog Output Latency	1.5 ms

AEC8

AEC Number of Channels	8 Channels
AEC Tail Length	512 ms - suitable for medium rooms
AEC Convergence Rate	Typically > 90 dB/sec
AEC Latency	16 mS
AEC Processors	ADI SHARC 21569@1GHz

Rear View



Control Software

VoxControl+ is our dedicated configuration software, available for free download from our official website. Designed with a user-friendly interface, it allows fine-tunusers to easily tailor the matrix settings to match the specific needs of any installation. With this software, you can e a wide range of parameters, including:

- Input gain
- Expander
- Compressor & Limiter
- Auto Gain Control (AGC)
- Equalizer
- Figure Balancer
- Active Noise Control (ANC)
- Feedback (AFC)
- Noise gate
- Ducker
- SPL
- Share AM (Automixer)
- Echo Canceller (AEC)
- Camera Tracking
- Noise Suppresion (ANS)
- Matrix
- Low & High Pass filters
- Delayer
- Output